Transmission Aspects of Data Transmission Service Using Private Line Voice Telephone Channels

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An exploration is reported of the possibilities of a moderately fast data transmission system to use private line message facilities. A comparatively conventional system was desired to permit expeditious application. An auxiliary "word start" signal was necessary for the system considered.

Transmission characteristics of a number of arrangements were examined. These included several exploratory AM vestigial sideband systems (using a spectrum similar to telephotography), double sideband AM systems, various telegraph and other multichannel systems.

It was concluded that the 1600 bits per second usable over the AM vestigial sideband arrangement was about as fast as could be expected with the data system contemplated. This transmission is not as "rugged," with respect to impulse noise and sudden level changes, as other slower arrangements, but it is expected to be satisfactory. It will require delay correction, and simple methods are considered for carrying this out.

I. INTRODUCTION

The Bell System has been approached on a number of occasions in regard to the transmission of computing machine and similar data over its telephone circuits. This has reached the point where specific possibilities for private line data transmission have been given serious consideration.

The telephone network was developed for speech transmission, and its characteristics were designed to fit that objective. Hence, it is recognized that the use of it for a distinctly different purpose, such as data transmission, may impose compromises both in the medium and in the special service contemplated.

A short time ago the authors were assigned the problem of examining possibilities for such an adaptation aimed at high speed, and exploring the nature of the transmission compromises that might he needed. As a result, a variety of data transmission systems in different stages of development have heen investigated including some telegraph systems. Certain conclusions have been reached regarding their suitability for use over private line telephone facilities. (By "private line" facilities are meant facilities leased to the subscriber on a more or less permanent private hasis, and that are not set up by operators at a telephone central office switchhoard.)

These conclusions are summarized below. In the later text there is included the background material for the conclusions which includes a brief characterization of the facilities. Finally there is included a discussion of the line treatment which may be needed for the best transmission.

It should he emphasized that not all possible data systems for use over telephone circuits are covered. The problem considered covers particularly some recently proposed applications, for which the need of a relatively high hit rate is important. Also, only the more promising comparatively conventional systems, which have been relatively well tested and can be readily applied, are considered. More radical designs are conceivable hut they would require more extensive investigation before conclusions could be reached concerning them. It is clear also that the designs involved in the choice of a system are determined by the type of service it is to provide.

1.1 *Conclusions*

It is concluded that about the fastest transmission of data which can be accomplished with the present art over message-type telephone facilities is obtainable with an amplitude modulated vestigial sideband system. Such a system will be presented in some greater detail below. Its frequency spectrum is similar to that of a telephotograph signal[2] and a number of the transmission prohlems involved are the same.

This system will provide about 1600 binary digits of data (or "bits") per second, hut it requires some special selection and considerable treatment of many types of circuits. This treatment is necessary to reduce noise, particularly of the impulsive type, and to correct for delay distortion.

Such a system is therefore considered suitable where a high bit rate is essential. It will be developed in the text that beyond the matter of the delay correction and other treatment of circuits required, the vestigial sideband process imposes a certain signal-to-noise penalty. A further signal-to-noise penalty comes from the method of multiplexing an auxiliary channel needed for word start indications.

Where a lower bit rate is acceptable, a 750 bit per second amplitude-modulated, double-sideband system which is now available, and has been tested extensively, is somewhat more rugged. It permits operation over the great majority of telephone facilities and will also be described in more detail.

The most rugged system considered uses frequency-shift transmission. This form of transmission has been used extensively for some time on a multiple-channel, voice-frequency telegraph basis in which each channel is capable of 46 to 74 bits per second. When used in this form to handle high speed data signals, it requires relatively complicated terminal devices because the several channels have to be merged to provide one high speed data system. However, when frequency shift is used as a single channel over an untreated telephone message facility the system promises to be relatively simple and to give a total possibility up to 1,200 bits per second according to the type of facility.

1.2 *Summary Table*

These findings are concisely grouped in a summary table, Table I. These entries are based upon present knowledge and are believed to be reasonably accurate, although estimates regarding impulse noise need more extensive checking, particularly in the case of the broad-band, frequency-shift system.

The table compares relative estimated performances of three broad-band systems among themselves, and with a subdivided channel (or telegraph) system. The performances considered cover the effects of noise and delay distortion, and the bit rates considered achievable in a 1,000-cycle band and in a 300- to 2,800-cycle telephone band. Some crude approximations covering speech are also given as a matter of interest.

The noise performance is given in terms of relative total power capacity required in the line for a given error rate in the presence of a given noise. The double sideband system is taken as reference. Allowance is made in the multiple channel or telegrapb system for occasional peaking caused by temporary unfavorable phasing. In this part of the comparison a 12 channel system has been assumed. This is about as many channels as can be used on a telephone facility that has heavy impulsive noise.

The delay distortion figures represent some present ideas on good engineering design in the allowable impairment of signal-to-noise ratio.

TABLE I — SUMMARY OF VARIOUS DATA SYSTEM CHARACTERISTICS

Use in Band	Relative Peak Power Required in Line for Equal Performance* in Presence of Noise, db			Approx. Max. Delay Distortion in Millisec. Allowable in Band (for 3 db noise penalty)		Approx. Bits per Second in 1000 Cycle Band (25-db S/N Random Noise for DSB)	Approx. Max. Bits per Second in Telephone Facility 300-2,800 Cycles
	SF	Random	Impulse	Within One Channel	Over 1000 Cycle Band		
Broadband VSB	+6	+6	+6	±0.4	±0.4	1000	1600**
Broadband DSB	0	0	0	±0.55	±0.55	700	800 to 1400**
Broadband FS	−3	−3	−7†	±0.5	±0.5	650	750 to 1200**
Telegraph Comparison							
43A1 Channel	+10†† (←12 Channels→)	0††	−10††	±5	±60	450 (6 Channels)	1100 (15 Channels)
Speech Comparison							
Speech	+10	+10	+10	±10			50

* High grade performance; i.e., less than 1 error per 100,000 bits.
** High figure assumes accurate delay correction and control of nonlinearity.
† Depends on precise line-up of filter and carrier.
†† Allowance for peak factor of 12 frequencies.
VSB = Vestigial sideband; DSB = Double sideband; FS = frequency shift.

This impairment is here assumed to be about 3 db. A rather larger impairment has on occasion been assumed by other authors.[3] In the case of the telegraph system some arrangements for merging the signals in parallel channels into a single high speed channel require a certain exactness in timing correlation. The need for this may be overcome by the use of a small amount of data storage in the receiver of each channel. The delay distortion figures quoted in the table assume no need for this timing exactness. As in the other part of the comparison twelve channels are assumed.

The last two columns indicate the bit rate that can be expected of the various systems, first per 1,000 cycles of band, and second for a telephone facility of somewhat narrow (but frequently encountered) bandwidth. Some of these figures assume a careful control of delay distortion and of nonlinear distortion. In the 1,000-cycle band only six channels of the telegraph assumed may be accommodated. In a 2,500-cycle band the number can be extended to 15. The band that can actually be used for telegraph, over a given facility, depends upon the nature of that facility.

Some comparison data are indicated for telephone speech. The bit rate given assumes particularly message communication and not finer shades of artistic expression. For this a collection of phonemes (or elementary sounds) in the fifties, needing six bits for identification, and a typical speech rate of eight phonemes per second, require 48 bits per second. A few bits are also needed for pitch indication, and the figure is rounded to 50.

The low bit rate obtained for telephone speech communication indicates considerable redundancy in the speech signal sent over the telephone channel. This suggests why substantial transmission impairments can be tolerated without destroying the intelligibility of speech, as compared with telegraph or data signals.

1.3 *Nature of Study*

The procedure followed has consisted of examining systems of binary or similar signal transmission which appeared suitable for the sending of data. The systems studied are listed here, and some description of them is given later in the text. As noted, part of the information has come from outside the Bell System.

1. Exploratory 1,650 bit per second vestigial sideband system studied at Bell Telephone Laboratories.
2. Exploratory 1,600 bit per second vestigial sideband system studied at Lincoln Laboratory of M.I.T.[1]
3. Exploratory 750 bit per second double sideband system, of general type reported by Horton and Vaughan.[3]
4. Voice frequency telegraph channels.[5]
5. "Polytonic" signaling system reported by Lovell, McGuigan and Murphy.[4]

The transmission problem of applying these various systems to the various types of telephone message facilities employed for private line service has been considered. These are also listed, and a brief characterization of them is given in the text.

1. Voice frequency circuits, over cable and open wire.
2. Type-C carrier circuits for open wire.[16]
3. Type-N carrier circuits for cable.[6]
4. Broad-band carrier systems[15] using A channel banks for paired cable, coaxial cable, open wire, and microwave radio.
5. Other broad-band carrier systems.

1.4 *Outline of Paper*

As a result of the study, some recommendations were made. The discussion of this material is presented in Section II. It covers first the sort of service, with comparatively high bit rate, for which there has been user demand, and which can be furnished over private line facilities. Secondly, the recommendations cover the broad features of the signal characteristics which appear promising for use over such facilities. These recommendations are at the present time evolving into an exploratory system. The recommendations themselves, together with some general remarks, are presented in Sections 2.0 and 2.1. The background material on which these were based, and which covers consideration of the five systems which have been listed, is presented in Sections 2.2 to 2.5.

The discussion on the nature of the problems involved in transmission of the signals over the telephone plant to be used (which above has been listed in five categories) is finally covered in Section III.

II. SYSTEMS OF BINARY SIGNAL TRANSMISSION

2.0 *General Remarks*

There are a number of arrangements in the present art, both experimental and commercial, which are capable of sending digital data information under something like the conditions required for the service considered. Study of these has led to a set of recommendations which are outlined below. The specific arrangements are then discussed in more detail. The description covers only the essential transmission characteristics of the arrangments.

The arrangments generally divide into two groups, those which are essentially short-pulse single channel (though some of these may include a slow auxiliary channel), and those which use frequency division multiplex channels and therefore employ longer individual pulses. One important advantage of the multiple channel systems is noted in Section 3.3 as consisting of an increased immunity against impulse noise.

The single channel group shows much similarity among the systems. The principal difference is that of bit rate. The faster systems use vestigial sideband transmission, at the expense of a certain increase in vulnerability to noise as compared with those which use double sideband transmission, and also at the expense of a more general need for delay distortion correction because of the speed.

The systems in this group which use vestigial sideband are expected to be able to operate, with a very low error rate (some 1 error per 100,000 bits), over telephotograph type circuits, which employ delay distortion correction. This is a "normal" condition error rate, and does not include abnormal circuit conditions, such as static, trouble conditions, etc. It also assumes a more or less even error distribution.

One of these systems gives very desirable word start indications by using an additional level of signal. "Words" are groups of signal elements of fixed total number each. A distinctive separation signal greatly simplifies their recognition at the receiver, particularly after short line interruptions.

As the price, however, of both the speed and the third level signal indication, this system is more vulnerable than the others to impulse noise, of the kind encountered in hitherto installed N1 carrier and open-wire circuits, which have not been treated for impulse noise. It is also more vulnerable than the other systems to sudden level changes in the received signal.

The multiple channel group is generally characterized by a lower bit rate. Of all of them only one, the frequency-shift carrier telegraph, shows capability of consistent operation over untreated N1 carrier. On this type of carrier the frequency-shift telegraph permits a bit rate of the order of only half that obtainable with the vestigial-sideband systems. This telegraph system performs much better over other types of telephone facilities but even then its bit rate is only about three fourths that desired.

2.1 *Digital Data Transmission Service*

2.1.1 *Some Desirable Major Requirements*

1. Transmission of a maximum of 1,600 bits per second with an error rate not to exceed one in every 100,000 bits (or once per minute).

2. Applicability to most telephone facilities for private line use. Some selection of circuits and some treatment of those selected will be acceptable. The circuits carrying data are to be one-way terminal circuits only (i.e., not to be linked in tandem). It is expected that for the bulk of the service the circuits would not be likely to run over some 200 to 300 miles in length, but a small number of 3,000-mile circuits is considered possible.

2.1.2 *Characteristics of the System*

The system which has been considered and recommended as promising for the service outlined above has the following essential characteristics:

1. Carrier at 2,000 cycles, with vestigial band extending up to some 2,400 cycles. Lower nominal effective band (half the bit rate in width)

extends down to 1,200 cycles, and roll-off band down to about 1,000 cycles. The frequency space above the vestigial band and below the roll-off band is essentially "dead" space, free of signal energy.

2. The signal comprises three components:

(a) *synchronization*, (or start), to indicate word separations,

(b) *data*, or the actual information, and

(c) *timing*, to mark out the successive bit intervals in the data.

The signal is represented in Fig. 1 as the envelope of a carrier. The synchronizing signal has an amplitude of one and a duration of one signal element. Data spacing signals have an amplitude of about 0.63 and a duration each of one signal element. Data marking signals have an amplitude of about 0.25 and also one signal element duration. It will be noted that this is an "upset" signal, in which spaces have more power than marks. The two signal elements immediately before, and again the two signal elements immediately after, the synchronization signal are always to be spacing. A timing signal is not actually sent, but instead the synchronizing signal is used to pull the phase of a highly

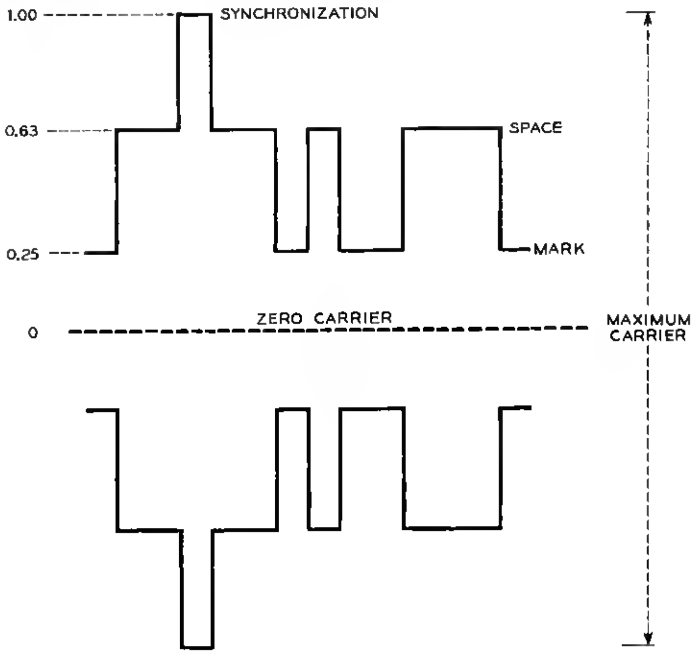

Fig. 1 — Multiple level signal.

accurate oscillator or "clock" at the receiver into the proper phase relation at the beginning of each word.

The phase of the oscillator is readjusted slightly as necessary at the beginning of each word but this adjustment is purposely made sluggish so that an occasional noise burst will not throw timing out badly. Thus the timing information is, in effect, by its repetitive characteristic, highly redundant. The receiving device is capable of getting in step without any manual adjustment but it may require as many as 10 words (or that number of synchronizing pulses) before it gets into exact phase, on an initial connection or after losing synchronization.

2.1.3 *Transmission Considerations*

Such a signal leads to the following transmission considerations:

1. The signal is rather similar in its spectrum to a telephotograph signal, and as a first order approximation requires about the same type of facility for its transmission.

2. In particular, the signal is expected to require something like the same over-all delay equalization as the telephotograph signal. A brief discussion of this point was given in a paper by one of the authors.[7] The conclusions there are that the telephotograph equalization limits of ± 0.4 times a signal element duration in envelope delay are generally reasonable, although these are probably overly severe with respect to very fine structure irregularities in the residual envelope delay characteristic. The formal limits which have been set on the envelope delay distortion for the 1,600 bit per second signal are ± 250 microseconds.

3. These limits constitute a rather less severe problem for the bulk of the expected circuits (200–300 miles) than they present for telephotograph circuits which are equalized for 3,000- to 5,000-mile lengths. For the small expected number of very long circuits the problem would of course be the same as for the telephotograph service.

4. The delay equalization problem requires special consideration in view of the nature of practices which have developed in the telephotograph art. The number of circuits which have been equipped for telephotography has in the past been rather small. The adjustment of the delay equalization has involved arithmetical calculation in the process of fitting various manufactured delay equalizer sections to the measured delay distortion. These methods are not generally suitable for large scale operations. It is believed that a process of equalization by prescription will be useable for the 200–300 mile circuits. This is discussed in more detail in Section 3.3.

5. So far, it has not been found expedient to transmit telephotograph signals generally over unmodified N1 carrier or other compandored

circuits, and the same problem is appearing with the proposed data signal. The problem is discussed in more detail in subsequent sections.

6. The specific signal suggested has the disadvantage that multiple levels must be discriminated by the receiver. This increases vulnerability to noise and level changes, as will now be discussed in some more detail.

The specific signal which has been suggested, as noted before, is outlined in Fig. 1. This signal comprises several levels. In the first place it includes a word start indication, on a separate level from the mark and space indications. In the second place the lowest amplitude level (normally a space, but a mark in the "upset" signal, as has been noted) is not made zero, but 0.25 the amplitude of the word start indication or "synchronizing" pulse. The need for this is explained in the discussion on vestigial sideband, below.

Consider for the moment a signal of only two levels; these can be taken as 1 volt and 0 volts respectively. The discrimination between them without error requires that an instantaneous noise pulse at this time be kept to less than $\frac{1}{2}$ volt. In these terms this represents an S/N ratio of 6 db.

The use of 0.25 volt minimum signal means that the amplitude range between maximum and minimum is reduced from 1 volt to $1 - 0.25 = 0.75$ volts. The maximum allowable noise pulse must be $0.75/2 = 0.375$ volts for this signal. This is an 8.8-db S/N ratio, as compared with the previous 6 db. It represents a handicap of approximately 3 db which must be accepted as part of the price of the increased bit transmission rate permitted by the use of vestigial sideband as compared with double sideband transmission.

An additional penalty comes from the use of three as against two signal levels. The spacing signal level is set at 0.625 volt, midway between 1 and 0.25 volt. Discrimination between synchronization and spacing signals can tolerate noise pulses of $(1 - 0.625)/2 = 0.1875$ volt. This amplitude is 15 db below synchronization level. Discrimination between spacing and marking signals tolerates maximum noise pulses of $(0.625 - 0.25)/2 = 0.1875$ volt. This is again 15 db below synchrouization level. Thus the signal tolerates a 15-db S/N ratio between synchronization level and the level of maximum noise pulses.

The difference between the approximate 9-db S/N for the two level vestigial sideband signal and the 15 db represents the 6-db handicap caused by the multiple level discrimination in the signal. This is the price paid for a distinctive word start indication.

The price also applies to sudden level changes. In a two level signal between 1 and 0 volt, a sudden drop of 6 db (without compensating change in the "slicing" or critical level) causes error.

In the two level signal between 1 and 0.25 volt, the permissible drop is reduced to 4 db (ratio of 0.625/1).

In the three level signal, the permissible drop is reduced to 1.8 db (ratio of (0.625 + 0.1875)/1).

Automatic gain control and corresponding adjustment of the slicing level ameliorate these conditions to some extent, but the problem is still a serious one; a sudden rise is also serious.

The use of a compandor in the transmission facility exaggerates the situation. Some discussion of the action of a compandor to improve the signal-to-noise ratio for speech is given in Section 3.1. For the moment it can be said that, at the transmitting end, the compandor compresses the range of amplitudes in the speech signal at approximately a syllabic rate. At the receiving end an expansion restores the original amplitude relationships in a complementary manner.

The compression and expansion are matched almost perfectly as far as the human ear is concerned. However data transmission is more vulnerable to short time level irregularities. This allows small imperfections in the amplitude restoration to impose a further penalty in the form of error hazards.

2.1.4 *Other Considerations*

Present ideas call for the acceptance of the signal by the telephone company, and redelivery to the subscriber not in the form indicated by Fig. 1, but in the form of the three separate components listed in 2.1.2. This presents some problems regarding the transmission of such signals over the connecting loops between the subscriber and the telephone office. These are not, however, germane to the general transmission problem and will not be considered further here.

2.2 *Bell Telephone Laboratories and Lincoln Laboratory Vestigial Sideband Systems*

The first of these represents an unpublished exploration, particularly by C. B. H. Feldman and A. C. Norwine, of the possibilities of transmitting moderately high speed data pulses over telephone facilities.

The exploratory system used a carrier at 2,200 cycles; a vestigial band extended up to 2,600 cycles; the nominal effective band extended down to 1,375 cycles; and a roll-off band continued down to 1,100 cycles. In order to reduce the quadrature component resulting from the vestigial sideband the spacing signal was made equal to one-third the marking signal amplitude. The receiver used AVC both for amplitude control and space-to-mark slicing level adjustment.

The timing of the receiver sampling instants to determine mark and space indications was determined by a flywheel circuit which operated from space-to-mark and mark-to-space transitions in the signal. This is similar in principle to the old Baudot quadruplex arrangements in telegraphy. It was a synchronous system and required a dummy signal for a lining-up period previous to the actual transmission of information. The lining-up was automatic but required some 15 to 50 milliseconds (25 to 80 signal elements).

A number of experiments with the system were made on actual lines. Most of these showed successful transmission, though the error rates were not measured quantitatively. The test showed the signal margin against error on a cathode ray oscilloscope. The wave trace indicated displacements both in the signal amplitude and in the timing of the sampling instant. This margin has been found to correlate reasonably well (in an inverse relationship) with the calculated delay distortions of the circuits used. It also corresponds reasonably well to theoretical expectations.[7]

The system reported on by the Lincoln Laboratory of MIT[1] shows much general similarity to the above. Perhaps the most distinctive feature of difference is in the use of a word start indicating or synchronizing signal, in the form of the high level pulse discussed above and somewhat similar to that used in television for scanning-line synchronization.

2.3 *Double Sideband Systems*

A prototype of these systems has been described by Horton and Vaughan.[3] Several models have been derived from the prototype which differ from it, somewhat, in several respects. Large portions of the systems are not germane to the present discussion, and only a brief description will be given of the signals.

An outline of the signal spectrum for the most recent of these derived models is illustrated in Fig. 2. The main signal is handled on a carrier at 1,500 cycles. The bit rate is 750 per second, and the nominal effective band is shown as ±375 cycles. A schematic roll-off is indicated. The words in this system are of about 100-signal element length, of which 8 are used for synchronization. The synchronization pattern involves a 3-signal element ready pulse on the 600-cycle carrier, simultaneous with the first 3 of the 6-signal element marking pulse on the 1,500-cycle carrier. Following this comes a spacing bit, then another marking bit on the 1,500 cycles only. The next bit is the first information bit.

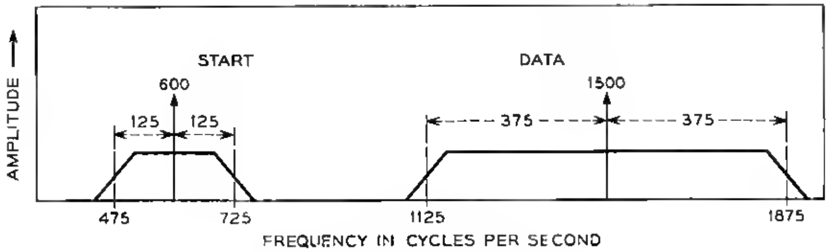

Fig. 2 — Double sideband signal with auxiliary start channel.

Marking consists of maximum carrier amplitude and spacing is zero carrier amplitude. The pulses are shaped both at the transmitting end and at the receiver before sampling.

The earlier prototype system as described by Horton and Vaughan was tested over a variety of telephone facilities (not including N1 carrier) running up to over 12,000 miles in length and found to be quite rugged. For reasons which are to be discussed later, the use of these systems over compandored circuits presents certain extra transmission problems regarding noise and level changes.

2.4 *Voice Frequency (VF) Carrier Telegraph*

The opposite extreme to the use of the telephone facility as a single band, to carry short duration pulses, is to subdivide the band into a large number of subchannels, each using longer duration pulses. As already noted, this has advantages against impulse noise, and also delay distortion.

There is available for this use the VF telegraph system.[5] In an AM form (40C1) this subdivides the space into 18 telegraph channels, which can each carry 100 words per minute (or 74 bits per second). A frequency shift form (43A1) is available, to give 17 channels, each to carry 74 bits per second.

The 18 channels use a band of 200 to 3,200 cycles in the telephone facility, and the lowest channel permits only a lower word speed. The 17 channels occupy the band from 350 to 3,200 cycles.

It has been mentioned that the telephone channels which provide the most serious problem for data transmission are those using compandors. The untreated N1 carrier is a principal example of such. Further, the type of circuits which use compandors are apt to be placed in plant which is relatively exposed to impulsive noise.

The principal interest in the telegraph channels, therefore, is to examine how they fare in application over untreated N1 carrier. There bave been some studies of this point. Tbe general conclusion, to be elaborated below, is tbat although the frequency subdivision (and also the frequency shift) helps against impulsive uoise, fewer telegraph channels may be used than over good non-compandored facilities; and at best, the transmission is accompanied by more distortion than expected in a telegraph link of the highest grade. However, a serious possibility for data transmission is indicated.

2.4.1 *Telegraph Tests on N1 Carrier*

Tests on this subject have been carried out by S. I. Cory, J. M. Fraser and others and reported in unpublished memoranda. Before presenting the results of the tests, some background is necessary on tbe terms in which the results are reported.

The performance is usually evaluated in tests of this kind in terms of a "maximum checkable" telegrapb distortion over a short time (about 5 minutes). The "telegraph distortion" represents tbe displacement from correct timing, of received signal transitions, after the initial mark-to-space transition in tbe "start" element. These displacements are measured in percentages of the signal element duration. By "maximum checkable" distortion is meant the maximum such displacement that is consistently reproduced in repetitions of the short testing period. This is somewhat larger than tbe root-mean-square distortion. A larger displacement is, of course, obtainable over a longer testing period. Although tbis measure of performance has had long use in the telegraph art, other measures are perhaps more readily grasped by and probably of more value to the data transmission engineer. Such a measure, for example, is the error rate.

The error rate may be estimated through the use of the telegraph transmission coefficient.[8] This is a figure which has been designed by telegraph engineers to indicate the performance of a telegraph circuit, particularly when it is made up of several sections. It is more or less proportional to the square of the distortion, and has the property that, when carefully chosen, it can be added for circuits connected in tandem. A small coefficient thus characterizes good transmission, and correspondingly, a large coefficient characterizes poor transmission.

The correlation between peak distortion over 5-minute intervals and error rate, through the telegraph transmission coefficient, is indicated in Reference 8 and Table II. It is to be understood that the correlation is only a rough one. Particularly at the two extreme ends of the scale, the entries in the table can serve only as a general guide to the performance, and the specific numbers are not to be taken too literally.

TABLE II — CHARACTERIZATION OF TELEGRAPH DISTORTION AND AND ERROR RATES BY TELEGRAPH TRANSMISSION COEFFICIENT

Distortion		Transmission Coefficient	Errors	
RMS	5 min. peak		1 in n characters	1 in m bits
			(n)	(m)
13.9	30	30	40	2.9×10^2
12.6	27	25	87	6.2×10^2
11.2	24	20	2.5×10^2	1.8×10^3
9.8	21	15	1.5×10^3	1.1×10^4
8.0	17	10	4.4×10^5	3.2×10^5
5.6	12	5	10^9	7.4×10^9
4.3	9	3	10^{12}	7.4×10^{12}
2.5	5	1	—	—

A very brief summary of some of the experiments in the use of VF telegraph over untreated N carrier is given in Table III. This portion of the results covers N1 circuits with compandors which have slightly more noise than the objective which is set for the telephone use of such circuits. The noise was 28 dba at the zero transmission level point, as against an objective of 26 dba at that point. It is noted in Section 3.3.2 that the measurement of noise in these terms is not altogether reliable in the evaluation of its effects on transmission systems that use pulses. Thus, these experiments must be considered as giving only a general indication of the situation.

TABLE III — SUMMARY OF TELEGRAPH PEFORMANCE OVER NOISY N-I CARRIER LINK

	40C1 (AM)	43A1 (FS)	
1. Number of channels	6	12	
2. Frequency space used	1020	2040 cycles	
3. Words per minute	75	75	100
4. Total bits per second	342	684	888
Average Channel			
5. Peak distortion	16	8	18 per cent
6. Estimated Transmission coeff.	9	2.5	11.5
7. Estimated errors, 1 in	10^6	10^{14}	10^5 bits
Worst Channel			
8. Peak distortion	25	17	22 per cent
9. Estimated transmission coeff.	21	10	17
10. Estimated errors, 1 in	1.5×10^3	4×10^5	5×10^3 bits

The tabulation is first given for an average channel. The performance of the worst channel has, however, also been included to give an indication of its contribution to over-all operation.

Many of the N1 carrier telephone circuits in the plant show lower noise than the objective, and to this extent Table III is somewhat pessimistic. Also some of the tests have shown, particularly for AM, that the performance is somewhat improved by removing the compandors. Thus, allowing for both points, better performance can be expected from the average N1 circuit (less than 200 miles) in the plant. The transmission coefficient of 11.5 listed for Item 6 at 100 words per minute might go down to say 9. At 75 words per minute with FS or AM, it might not be over 4.5. It is clear, of course, that with further modifications of the N1 channels, such as to reduce noise exposures, better performance could be obtained.

The broad conclusions that can be derived from these considerations are:

1. The subdivision of the frequency band into telegraph channels, and the use of FS, permit a workable system to be operated over a compandored facility like the N1 carrier without modification. This occurs even when the latter has noise up to and a little over the telephone objective.

2. This workable system under such noisy conditions transmits up to some 350 bits per second with AM, and some 800 bits per second with FS. It is accomplished with an error rate of the order that has been implied for data transmission, even in the worst channel.

3. There is a relatively wide range of performance of the system over different N1 circuits, and the average performance is sensibly better than that under the limiting conditions which have been considered.

2.4.2 *Distribution of Signal in Allocated Bandwidth*

A more extensive discussion of the use of bandwidth is given in Section 3.1, below. However, a few specific points are appropriate here on the band use in telegraph channels.

The spectrum of the original voice frequency telegraph system, based

Table IV — Use of Frequency Spectrum in Telegraph Channel

	AM	FS
1. Channel spacing	170	170 cycles
2. Nominal effective bands	74	74
3. Roll-off band (both sides)	37	26
4. FM swing	0	70
5. Guard band (both sides)	59	0

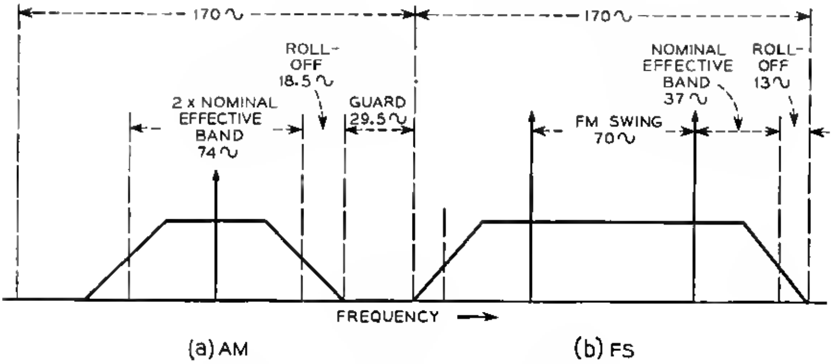

Fig. 3 — Utilization of telegraph channels.

on 170 cycles between carriers, was conservatively developed for the 60 word per minute speed of the time. The 100 word per minute speed has used up some of this conservatism. The use of frequency shift in the same channels has, however, used up the spectrum space even more.

An outline of the band allowances is given in Table IV, and illustrated in Fig. 3. Item 2 of the table is based on the 100 word per minute speed, using double sideband. On this basis, the number is equal to the number of bits per second. This is the minimum double sideband over which that number of bits can be transmitted, according to the Nyquist theory. Each such sideband is sometimes called a "nominal effective band." In practice various allowances are necessary over this minimum.

In the first place a roll-off is necessary because filters are not infinitely sharp, and in addition the nature of the modulation itself forms a roll-off. Roll-off also leads to a signal which is more free of overshoots and generally "cleaner" than when a sharp cutoff is used. Item 3 and Fig. 3(a) and (b) show an allowance for roll-off. For the AM case this amounts to half the nominal effective band. For the FS case there is not quite that much space available.

For the FS signal it is necessary to allow for the frequency swing as Item 4. For the 43A1 system this amounts to 70 cycles. In Fig. 3(b) the spectrum includes the region comprised by the FM swing, and upper and lower sidebands. The upper and lower sidebands as formed by the modulation of a random signal are shown extending respectively above and below the extremities of the swing (instead of only above and below a central carrier, as they would with AM).

A final allowance in Item 5 is a "guard band." This is taken to mean a region in which the signal energy is negligible, but at the same time a region in which no appreciable interference is tolerated from the adjacent band. The allowance for this is generous in the AM case. In the FS case the roll-off band of Item 3 tends to use up all the space not employed by the nominal effective band and the swing, and nothing is indicated as left for guard band. This is illustrated in Fig. 3(b) by the extremities of the roll-off band reaching the extremities of the 170-cycle spacing. It is to be recognized that the illustrations are diagrammatic. However, the extremities mentioned measure tbe bands occupied by power from random signal transitions between marking and spacing (as distinguished from mere mark-space reversals), analogous to the bands occupied by power from AM signal transitions. Comparison of Figs. 3(a) and 3(b) suggests how modulated signal components of significant intensity are displaced farther from the edges of the channel with FS than with AM.

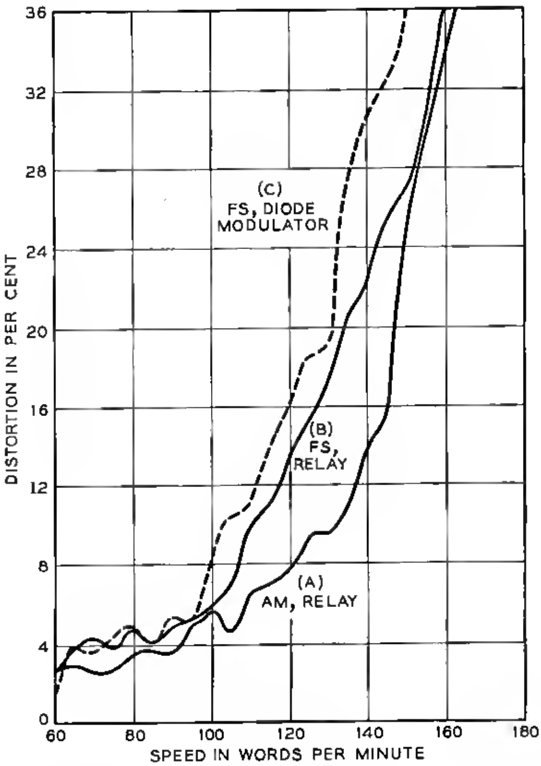

Fig. 4 — Experimental relation between telegraph distortion and speed, with fixed channel filters (after Jones and Pfleger[9]).

The conclusion is reached that there is hardly any excess conservatism in the 43A1 system. Some confirmation of this is indicated in a paper by Jones and Pfleger.[9] Fig. 4 reproduces some curves presented in that paper. The curve at (B) (from Fig. 5 of that paper) shows that the FS telegraph rises rapidly in distortion above the 100 word/min speed. The curve at (A) (taken from the same Fig. 5) for level-compensated AM shows a substantially broader and somewhat lower curve in this region. From curve (C) (taken from Fig. 3 of the paper) for diode modulated signals, it is seen that the sharp rise in distortion for FS is even more accentuated than for the relay modulator. This indicates how much more characteristic distortion FS exhibits than AM because of the sharper roll-off of its signal bands within the confines of the same 170-cycle channel spacings. Of course, as the word speed is raised beyond the practically usable values, either type of modulation leads to so much power in the filter cutoff regions that the characteristic distortions become more or less indistinguishable.

2.5 "*Polytonic*" *System*

This is a frequency discrimination system experimentally proposed for toll and local signaling.[4] It works on 5 channels at speeds of 100 and 300 decimal digits per second (or the equivalent of some 330 to slightly under 1,000 bits per second). It has some similarities to an earlier multi-frequency system,[10] but is faster.

The distinguishing characteristic of the polytonic system lies in the mathematical theory which has been followed to reduce interchannel interference. This analysis makes use of the theory of orthogonal functions, and is similar to that used in the computation of Fourier components. The mathematical analysis leads to an ideal receiver design for minimum interchannel interference. The ideal detector in this receiver closely resembles the conventional homodyne detector. The detector actually used, however, represents a practical simplification of the latter. A complete description cannot be given here, but it may be noted that the theory leads to a need for synchronization of the signal elements in the five channels, and to the setting of an exact timing instant for the sampling of the received wave to obtain minimum interchannel interference. The indication for this instant is obtained from the use of a sharp wave-front pulse in the marking channels.

Tests with the 100 decimal digit per second system (100 decimal digits normally correspond to 332 binary digits) indicated that it gave good operation[4] over toll circuits of comparatively limited length (350 miles for four-wire voice frequency, 1,900 miles for K carrier). The higher speed system was designed only for local plant.

It is clear that this system has too low a bit rate for the application contemplated, even in the faster form. It is also not generally adaptable to the variety of circuit lengths which are expected to be encountered.

III. UTILIZATION OF THE TELEPHONE CHANNEL

The discussion herewith covers a broad examination of some major characteristics of telephone communications facilities, to evaluate their bearing on the choice of a system for the data transmission service outlined before. It is of course clear that different conclusions might be reached for other types of service.

The first item is an outline review of the different types of message telephone facilities in the plant. This is followed by an analysis of the different possibilities in the use of the frequency spectrum, of noise, and of delay distortion, in the application of the data signals.

3.1 *Telephone Facilities*

There is a rather wide variety of facilities to be found in the telephone plant, to be examined with respect to the factors that are germane to the present question.

The first of these factors is the frequency bandwidth capability of the facility. For message-type voice circuits, this is generally characterized as being three kilocycles (with the exception of "emergency banks," which are substantially narrower, and are not to be considered as useable for data transmission).[17] However, some of the telephone circuits in the plant, aside from the emergency banks, are also somewhat narrower than 3-kc, and in any case, not all the band is effectively usable for data transmission. As will be seen, the net available band is, in practice, about half of the 3kc.

Part of the reason that not all of the frequency band is effectively usable is that the circuit shows delay distortion. This tends to become large at both the lower and the upper edges of the band. Some details of the delay correction are discussed further below.

Another impairing factor in telephone facilities is the nonlinear distortion encountered. In voice frequency facilities this comes from amplifiers and loading coils, and increases progressively with circuit length. In carrier facilities the nonlinear distortion arises almost exclusively at carrier terminals, in the part of the circuit where the signal is at voice frequency. In such a case, the distortion increases with the number of times in the telephone facility that the signal is modulated down to voice frequency. Second order nonlinear distortion tends to develop modulation products in the lower portion of the transmission band which are a source of potential interference with the signal.

Another impairment encountered in carrier telephone channels is a slight frequency shift; that is, a 1,000 cycle input may appear at the output, say at 998 cycles. This occurs because modulator and demodulator frequencies are not identical. With independent oscillators on recent systems this shift may amount to some two cycles. With older systems it can run from 5 to 10 times as much. The effects of this shift are discussed below. The frequency shift may be avoided by working double or vestigial sideband and using an envelope detector, or in some carrier systems by locking the oscillators in a constant frequency network. This locking may or may not result in close phase synchronization of the carriers, depending on the method used to lock and the particular carrier system involved.

Still another factor is the use, or not, of "compandors." A compandor compresses the range of speech volume in the impressed line signal and correspondingly expands this range at the receiver. This raises the line signal level during periods of low speech power, and lowers it during periods of high speech power without, in principle, affecting the final received level. The effect is to reduce the final noise in periods of low speech power, and increase it during periods of high speech power. A listener is less perceptive to the noise during high speech power levels than low. By this means, it has been found that the telephone circuit can be engineered to some 23 db more noise (and also crosstalk and similar forms of interference) than it can without the use of a compandor.

In the case of data signals, however, the influence of noise in causing error is not very much different whether the signal is marking or spacing. Thus, there is no "compandor advantage"* (indeed there is a certain disadvantage as pointed out earlier), and facilities that have been engineered to be entirely satisfactory for voice transmission are effectively some 23 db more noisy for data transmission. As a practical matter it appears desirable to remove compandors from circuits used exclusively for data.

A short listing is presented here of the various types of message facili-

* Perhaps a simpler way to think of it is that all possible "compandor advantage" has already been obtained in a data system by using the best combination of amplitudes for mark, space, start, etc.

ties most frequently found in the telephone plant, and some comments are made on each.

3.1.1 *Voice Frequency Circuits*

There is a variety of open-wire facilities of this type. They are mostly short, and two-wire. Thus, repeaters can to advantage be turned one-way for data service. Delay correction is discussed later.

Voice frequency cable facilities over more than a very short distance are loaded. This gives appreciable delay distortion. The loading used is indicated by a letter denoting the spacing, followed by a number denoting the loading coil inductance. Thus "H-44" means 6,000-foot spacing, of 44-millihenry coils, and "B-88", 3,000-foot spacing of 88-millihenry coils. Conductor capacities range from 0.062 microfarads per mile for toll circuits, to 0.082 microfarads per mile or sometimes even higher for local circuits. This affects the delay distortion.

3.1.2 *Type-C Carrier Circuits*[16]

This is an open-wire three-channel system operating at different frequencies in opposite directions, over the same pair. Historically there has been a variety of C systems developed, but only the C-5 system exists in any extensive quantity. The upper frequency cutoff in the voice channel is well under 3 kc. The delay distortion varies widely with the specific channel and direction of transmission. There is a variety of channel frequency allocations, and the distortion varies with this also. The delay distortion over some channels increases rapidly above 2,400 cycles. The frequency shift discussed before may be as much as ±20 cycles.

3.1.3 *Type-N Carrier Circuits*[6]

This is a short-haul twelve-channel system for use over cables. Because of its economy it has been extensively introduced. Its principal characteristic, in the application of data circuits, is that it uses compandors. It therefore presents a noise problem. The delay distortion, introduced almost exclusively by the terminals is not excessive, and depends very little upon circuit length between the terminals. The N system uses double sideband transmission, and therefore exhibits no frequency shift between input and output signals.

3.1.4 *Broadband Carrier Systems Using A Channel Banks*[15]

There is a variety of carrier systems designed for paired cable, coaxial cable, open wire, and radio, that use a standard grouping of twelve channels with associated filters, known as an "A channel bank." The delay distortion in these associated filters constitutes nearly all of the distortion measurable over the complete system. These are single sideband systems, and unless the local modulator and demodulator carrier supplies are locked in a constant frequency network, frequency shifts of some ± 2 cycles may be expected between input and output.

For paired cables, these are known as K1 and K2 systems. For coaxial, they are L1 and L3, for open-wire, J, and for microwave radio, TD-2.

3.1.5 *Other Broadband Carrier Systems*

An O carrier system has been developed for open wire, and combinations of it are used with N for open-wire and carrier. Tbese are compandored systems.

3.2 *Use of Bandwidth*

This section examines the more important factors which affect the choice of how the available bandwidth of a facility is to be used, either in one band or a subdivided band.

3.2.1 *Baseband Transmission*

This is the simplest type of transmission. It is used in telegraph loops and other short distance telegraph transmission. A mark is indicated by placing marking voltage across the wire line, and a spaee by placing spacing voltage. In the simplest systems the latter is zero. In "polar" systems it is the negative of marking voltage.

The frequency spectrum of the signal runs down to and includes dc, as illustrated by the solid lines in (a) of Fig. 5.

With many transmission facilities it is difficult or impossible to transmit the dc; i.e., the circuit cuts off as is illustrated by the dotted lines. In such cases it is impossible to distinguish between a permanent mark and a permanent space.

Extra pulses can, however, be added to the signal to insure that marks or spaces are not permanent, but are relieved by the opposite signal in some maximum interval of time. In such cases the received signals can be clamped on mark or space signals and the opposite condition can be readily distinguished. This is sometimes called "dc restoration," and strictly speaking the system ceases to use baseband transmission. It may be designated as "modified baseband transmission." Methods other than clamping have been suggested for dc restoration.

Reverse pulses can be systematically inserted after each mark or space pulse, according to various patterns.[11] Two suggested are "dipulse" and "dicode" pulses. Such signals approach carrier signals, which are discussed below.

The principal weakness of baseband transmission appears when it is sent over C carrier or other single sideband telephone facilities, where the recovered signal may vary in frequency from that sent. This causes a distortion of the received pulse whicb makes it difficult to recognize.

An analysis of this point is given in Appendix I. It is concluded that while tbere may be long range possibilities in baseband transmission, it requires more study, and it will not be considered further in this paper.

3.2.2 *AM Carrier Transmission*

The simplest of this type is double sideband transmission, as illustrated by the full line of Fig. 5(b).

A comparison of the susceptibility to noise of this arrangement, with that of baseband transmission, is considered in the next section.

A further consideration required is susceptibility to nonlinearity in the facility. Second order modulation leads among otber tbings to a rectification of tbe signal back to baseband. This is indicated by the dotted lines in Fig. 5(b). After such rectification of the signal by the facility, it is impossible to separate any overlapping portions of the signal between the baseband and lower sideband. Some overlap is shown. This interference was first considered in telephotography[12] and is known as "Kendall effect."

Tbe possibility of Kendall effect may be eliminated insofar as second order modulation is concerned by moving the carrier frequency higb

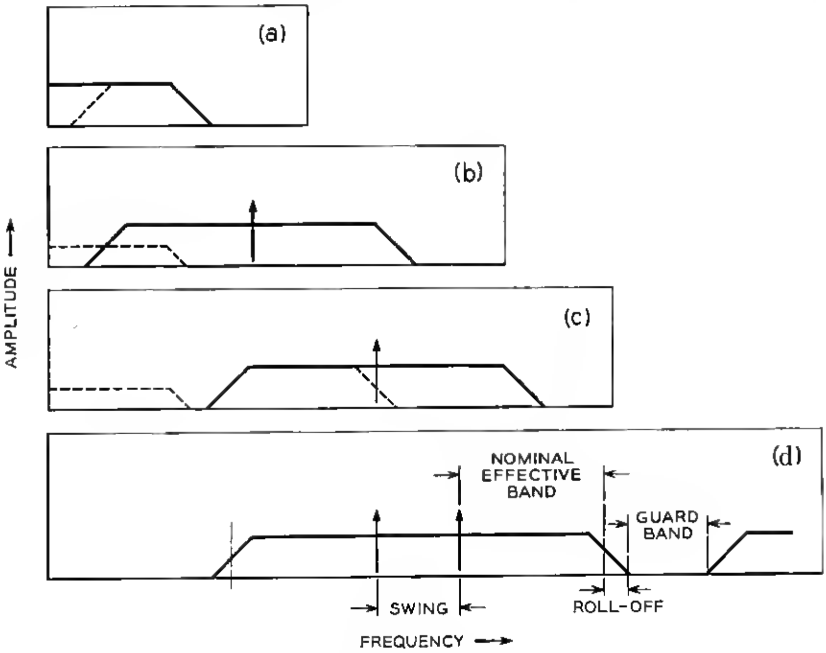

Fig. 5 — Spectra of signals with various modulations.

enough to prevent such an overlap. This is indicated in Fig. 5(c). It does not prevent third order modulation effects.

It has been found necessary to allocate frequency bands thus to avoid the overlap discussed in the transmission of high grade telephotography over telephone type facilities. It has also been noted that allocation with such overlap was undesirable in some data transmission experience. However, the question has not been resolved in complete detail. For the data service under consideration, which is expected to show a very low error rate, it is deemed conservative to allocate bands without the overlap.

This conclusion then leads to wasting a certain part of the lower frequency range. It is still possible, however, to use this range for an auxiliary signal, as in the system illustrated in Fig. 2, if the auxiliary signal occurs only during word starts.

The double sideband frequency range, as indicated in Fig. 5(b) or 5(c), is about twice the baseband range of Fig. 5(a). It is possible to reduce this extension by cutting down one of the sidebands to a "vestige" of itself and sending carrier at a reduced level, as indicated by the diagonal dotted line about the carrier in Fig. 5(c). This was proposed by Nyquist.[13] It is done at the expense of an increased vulnerability to noise, which in total amounts to some 5 or 6 db in certain typical cases.[7, 11] In Section 2.1.3 discussion was given to account for 3 db of this. In the references cited herewith it is noted that vestigial sideband transmission is accompanied by an interfering component (called a "quadrature component") which accounts for the other 2 or 3 db.

3.2.3 *FM Carrier Transmission*

Certain additional immunity to noise is gained by the use of frequency modulation (or "frequency shift") of the carrier. The immunity which can be obtained against impulse noise can be even greater than that against random noise, provided that the receiver is precisely tuned. This was noted in Section 2.4 in connection with voice frequency telegraph.

The noise immunity obtained from FS is in part due to the use of a higher average power and is at the expense of a wider frequency band as illustrated in Fig. 5(d). In addition to all the band that is used for a double sideband system, a space must be allowed for the swing. FS is also much less vulnerable to sudden level changes than DSB and thus may well be preferable to DSB for medium bit-rate service. As shown in Table I, these advantages can be obtained with only a small sacrifice of hit rate compared to DSB for equal bandwidths.

So far the single channel broadband FS system has not been generally used over wire circuits. Hence its exact performance, particularly with impulse noise, is only estimated.

On occasion not all of the band illustrated in Fig. 5(d) is allowed for. This leads to an increase in distortion of the signal, which has some similarity to a very close-in echo. It uses up some of the additional noise immunity provided by the FM, as an engineering compromise.

Another direction along which the FM system may be practically extended is to use four instead of two (marking and spacing) frequencies. This would double the bit capacity at the expense of only a moderate widening of the frequency band and somewhat tighter requirements on noise and delay distortion (but not of level regulation, which would be required for a similar extension of the AM signal).

3.2.4 *Multichannel Systems*

It is possible to divide up the entire frequency band available into a number of separate channels and use any one of the various carrier systems which bave been described, in each individual channel. This may be done because the nature of the information transmitted may be better adapted to the narrow channel, as in conventional telegraph. It permits certain elements of flexibility in layout, and offers certain noise advantages (and also disadvantages) as discussed in the next section.

In an idealized way one can proportion the various allowances for nominal effective band, roll-off, guard band, and swing (FS) in the same proportion in which they would occur in a single broad channel over the whole facility. Thus no frequency space would be lost by the subdivision. In practice, however, subdivision usually does lead to some actual loss in the frequency space.

A significant limitation to frequency subdivision lies in nonlinearity of the facility. This leads to modulation products between the various channels, which interfere with other channels.

In the case of voice frequency telegraph and other multiple channel systems the modulation effects are mitigated by allocating the carriers at odd multiples of a basic frequency. That is, any given carrier f is set at $f = nk$, where n is odd and k is a basic figure. Then the three second order modulation products are

$$2f = 2nk,$$

$$f_1 + f_2 = (n_1 + n_2)\,k,$$

$$f_1 - f_2 = (n_1 - n_2)\,k.$$

TABLE V — USE OF TELEPHONE BAND BY VARIOUS DATA SYSTEMS

System	Modulation	Max. No. of Channels	Bits/Sec. per Channel	Total Bits/Sec.
1. Proposed	VSB	1	1600	1600
2. Exploratory	VSB	1	1650	1650
3. Exploratory	DSB	1	750	750
4. Polytonic toll	DSB	5	100	500*
4. Polytonic local	DSB	5	300	1500*
5. 40C1 Teg.	DSB	18†	74††	1332††
6. 43A1 Teg.	FS	17†	74††	1258††

* Not realizable with 2 out of 5 codes used.
† Not realizable over some facilities.
†† Based on 100 word per minute channel capability.

All these products are necessarily even multiples of k and therefore always fall half way between carriers. This allocation, however does not permit mitigation of third order modulation effects.

3.2.5 *Experience*

Table V reviews systems which have been discussed earlier in this paper, to indicate the extent to which they use a general telephone channel facility, in terms of the bit rate output. It is clear, of course, that the various systems are not engineered to the same conservatism. These differences have already been commented upon.

The general conclusion which one can reach from the table is that the use of the medium in a single channel gives possibilities of a higher bit rate than subdivision. However, it is to be kept in mind that the telegraph facilities are conservatively engineered. Further as noted, they can be used to the full extent indicated only over the broader band telephone channels. For example, the full 18 telegraph channels can not be used over a C carrier telephone channel.

3.3 *Noise*

A general theory regarding the influence of noise on digital systems was presented in 1948 by Oliver, Pierce and Shannon.[14] This was discussed further at a symposium.[7] Several additional points are considered here, relating to the effect of noise on a data transmission service of the type contemplated.

3.3.1 *Effect of Channel Subdivision on Vulnerability to Noise*

This has been suggested earlier at several points. A more detailed discussion is given of the effects in Appendix II.

The conclusions reached there are, broadly:

1. Channel subdivision has comparatively small effect on vulnera-

bility to random noise. The small effect which does occur is a disadvantage which can run up to some 3 or 4 db for ten or twelve channels, for the multichannel as compared to the single channel system.

2. Channel subdivision is advantageous over single channel use, with regard to vulnerability to impulse noise.

3. Channel subdivision is disadvantageous compared to single channel use, with regard to vulnerability to single frequency noise.

3.3.2 *Noise Effects in Vestigial Sideband System*

Some brief discussion of the noise problems in the vestigial sideband system under consideration has already been presented in Sections 2.1.3 and 3.1. That such problems may be important in the use of actual telephone facilities has generally been checked by some unpublished tests carried out by J. Mallett, of Bell Telephone Laboratories.

The noise problem is most serious in the applicatiou of data transmission over N1 carrier. It is particularly significant for the N1 installations previous to the most recent. The most recent installations are engineered with distinctly more conservatism in regard to noise performance. As is noted in Section 3.1, the principal characteristic of the N1 system that affects this application is that it uses a compandor and that its design for telephone use assumes a reduction of the noise by this compandor. The reduction then is not realized for data signals. A second characteristic is that N1 channels are exposed to impulse noise. The channels are of course designed to limit such noise to the extent that it will not sensibly impair telephone speech. But short data pulses are more vulnerable to the impulse noise than speech. The 2B noise meter, which is normally used for telephone noise measurements, does not read sharp noise impulses according to their effect on data signals, and other methods of measurement have been explored.

A summary of some of Mallett's results is plotted in Fig. 6. Here the reading on the 2B meter is compared with that on a level distribution recorder which records peaks of 1 millisecond or longer. The "one per cent" point is noted, which means that one per cent of the one second intervals in the period of measurement contained one or more peaks (of 1 millisecond or longer duration) of the amplitude indicated. This is found convenient as a measure of the error frequency tolerated in the system considered (one in 10^5 bits).

The heavy dots indicate the correlation between the two sets of noise readings on idle tested channels of an operating N1 system (other channels in the system being busy due to normal use). All channels had compandors in. Dots that mark the boundaries of a group of tests (usually a single system in the group) are connected by straight lines. The open

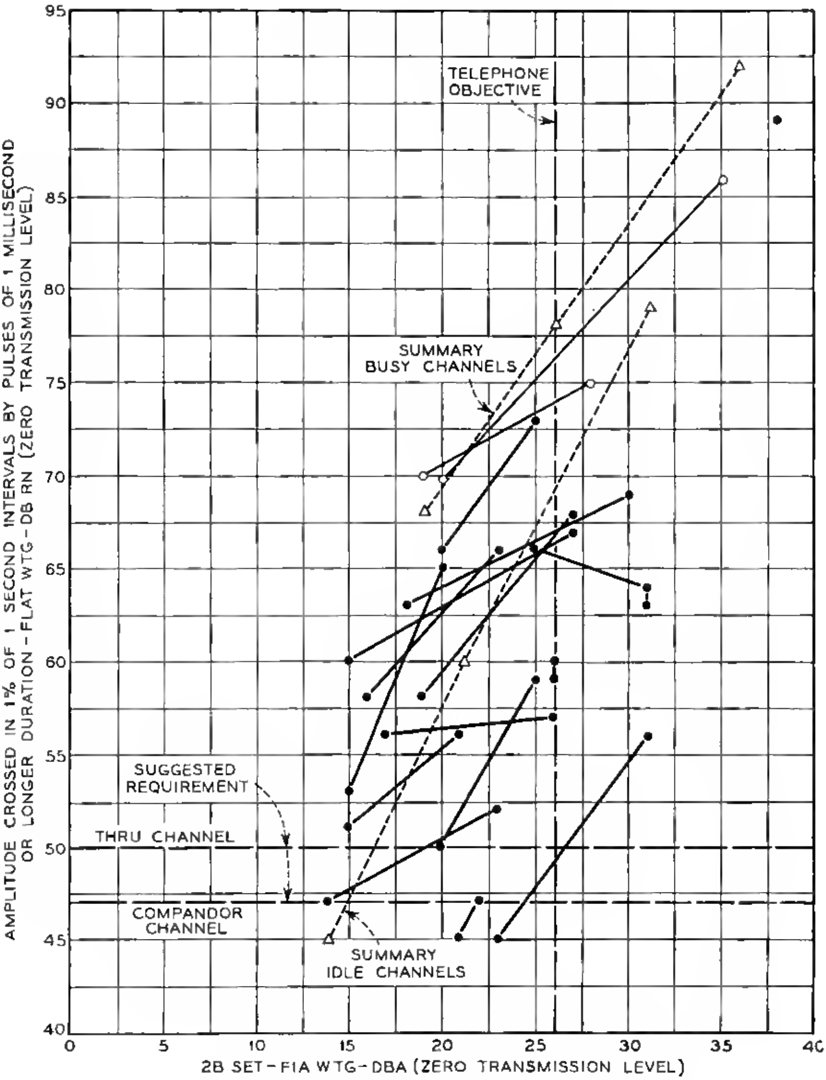

Fig. 6 — Noise measurements in N1 carrier facilities.

dots refer to an estimate (computed from the known properties of the compandor) of what the noise peak levels would have been, had the channel tested been busy with an operating data transmission signal (of the general type illustrated in Fig. 1). Under this condition the compandor setting would of course be different from that of an idle channel.

In such cases the 2B meter reading is also adjusted to the effective message circuit noise which would have existed in the presence of speech on the tested channel.

Open triangles connected by fine dotted lines indicate summary plots for the idle channels, and for the busy channels.

Examination of the plot shows that there is only a general correlation between the 2B set and level distribution recorder readings, as summarized by the dotted lines. Thus the 2B meter is not a reliable instrument to denote how noisy a circuit is for data transmission of the speed used in the proposed project.

The telephone objective for N1 circuits, as read on the 2B meter, is indicated by the vertical dotted line. This shows that most of the circuits (actually 55 out of 62) met the objective.

The suggested requirements for data are shown by the horizontal dotted lines. The "through" or noncompandored channel has a 3 db more lenient requirement (as indicated during some of the tests) than the compandored one. This results from the penalty mentioned earlier which compandors impose on in a data circuit, as compared with the 23 db or so advantage that it introduces for voice transmission. Only a few of the circuits measured (actually 10 out of 62) met the suggested requirements.

The principal conclusion reached from these measurements is that where used for a data service of the type considered, with vestigial sideband transmission, most of the hitherto installed N1 circuits (and probably other compandored circuits) will require modification to reduce noise exposure. Also noise measurement will be more complicated for such a service than for telephony.

3.4 *Envelope Delay Distortion*

Some simple theoretical considerations[7] have shown that the envelope delay distortion limits for a telephotograph circuit generally also hold for a data transmission circuit of the same speed. The principal difference is that less emphasis need be placed for data circuits upon fine structure deviations of the envelope delay as plotted on a frequency scale. In general, distortion of ± 0.4 signal element in the important part of the band has been found to give a signal-to-noise impairment of some 3 db in signal reception. This has been assumed here as a tentative engineering objective.

In aecordance with this, the envelope delay requirements for service with the vestigial sideband signal consideration have been set at not to exceed 500 microseconds (± 250 microseconds) between 1,000 and 2,500 cycles. This contains an element of conservatism inasmuch as the strict requirement is really fully implied only on the nominal effective band (1,200–2,000 cycles). The signal power is reduced in the roll-off and vestigial bands, respectively 1,000–1,200 and 2,000–2,500 cycles, and some corresponding liberality may be expected there.

The delay distortion constitutes a more serious problem with a faster system as compared with a slower one, in part because of the wider frequency band occupied by it, and in part because 0.4 signal element represents a more severe tolerance in microseconds for a shorter element than for a longer one. Consequently, the limits given represent about as severe tolerances as may be expected to be needed with the use of a telephone channel.

The distortions of various circuits have been considered to estimate the order of the problem involved in meeting the proposed requirements over links of 100 to 500 miles.

The following conclusions are reached first for the vestigial sideband signal, and after this for the slower systems.

3.4.1 *Facilities Requiring No Treatment*

As already noted, K2, L1, and L3 carrier, and TD-2 microwave, use "A" channel banks to separate the individual channels, and these give the dominant delay distortion. This amounts to a maximum of about 200, and a minimum of 150 microseconds, according to the exact combination of filters used. This figure is for one link of transmitter and receiver. A single section delay equilizer can cut the maximum residual to about 80 microseconds. It is concluded that these facilities present no important delay distortion problems. An N1 carrier link gives a maximum delay distortion of 220 microseconds, which can be reduced to 50 microseconds by one section of equalizer. This, then, also presents no serious problem.

3.4.2 *Facilities Treated by Simple Prescription*

The delay distortion of H-44 voice frequency cable in the 1,000- to 2,400-cycle range runs to slightly under 900 microseconds for 300 miles, if the cable is of standard toll capacitance (0.062 mf per mile), and to slightly under 2,000 microseconds if of higher local plant capacitance (0.084 mf per mile). The use of about one section of equalization per 100 miles reduces the residual to less than 330 microseconds for the low capacitance cable. For the higher capacitance cable, about three sections are needed per 110 miles. The J-2 carrier uses A channel banks, but has, in addition, directional separation filters at each repeater. This gives maximum and minimum distortions, respectively, for 100 miles, of slightly under 300 and slightly under 160 microseconds. The precise distortion in any given channel depends upon its proximity to the cut-off of the directional filter. For 500 miles the figures are slightly over 500 and slightly under 50 microseconds. With the same single section of equalization the maximum figures are reduced to about 100 microseconds for 100 miles, and about 300 microseconds for 500 miles. To carry out this equalization requires only rudimentary information on the general nature and correction of delay distortion. If moderate care is used in the prescription of equalization on a packaged basis no delay measurement of the circuit would in general be needed, though it is recognized that some difficult cases may arise.

3.4.3 *Facilities Requiring More Involved Prescription*

The delay distortion in C-5 carrier[16] is influenced to a dominating extent both by channel and directional separation filters. It varies in a complex fashion from channel to channel, and according to the direction of transmission. Its correction thus requires more involved prescription than is required for the other types of circuit. In some few cases measurement may be necessary. The distortion of H-88 voice-frequency cable runs from some 1,400 to over 3,000 microseconds per 100 miles according to capacitance. For 20 miles of H-174 toll cable the distortion is slightly under 1,400 microseconds, and its use is not contemplated.

3.4.4 *Data Systems Requiring No Corrections*

The delay distortion problem is practically non-existent for the slower systems. For the double sideband systems some delay correction may be needed if long heavily loaded circuits are used or perhaps for some other rare unfavorable situations, but otherwise no correction is necessary. No correction is needed for the telegraph systems.

APPENDIX 1 — BASEBAND SIGNAL DISTORTION CAUSED BY CARRIER FREQUENCY SHIFT

A simple analysis of the phenomenon may be considered. Let the voltage input, as in Fig. 7(a), be a raised cosine pulse between the angular arguments of $-\pi$ and $+\pi$. That is

$$V_i = 1 + \cos \Omega t, \qquad (1)$$

where Ω/π is the envelope frequency. When this is transmitted on the carrier, $\cos \omega t$, the carrier signal voltage is

$$V_c = (1 + \cos \Omega t) \cos \omega t, \qquad (2)$$

$$= \cos \omega t + \tfrac{1}{2} \cos (\omega - \Omega)t + \tfrac{1}{2} \cos (\omega + \Omega)t. \qquad (3)$$

When the carrier and one sideband are removed (say the lower sideband), V_c becomes (neglecting the factor $\frac{1}{2}$)

$$V_c = \cos(\omega + \Omega)t. \tag{4}$$

At the receiving end, V_c is modulated with a carrier which may momentarily differ in phase from the signal carrier by angle φ, giving

$$V_o = \cos(\omega + \Omega)t \cos(\omega t + \varphi), \tag{5}$$

$$= \tfrac{1}{2}\cos[(\omega + \Omega)t - (\omega t + \varphi)] + \tfrac{1}{2}\cos[(\omega + \Omega)t + (\omega t + \varphi)]. \tag{6}$$

The lower frequency part of V_o constitutes the recovered signal, and it is extracted by a filter that attenuates the higher frequency part. Thus, again neglecting the factor of $\frac{1}{2}$,

$$\begin{aligned} V_r &= \cos(\omega t + \Omega t - \omega t - \varphi), \\ V_r &= \cos(\Omega t - \varphi). \end{aligned} \tag{7}$$

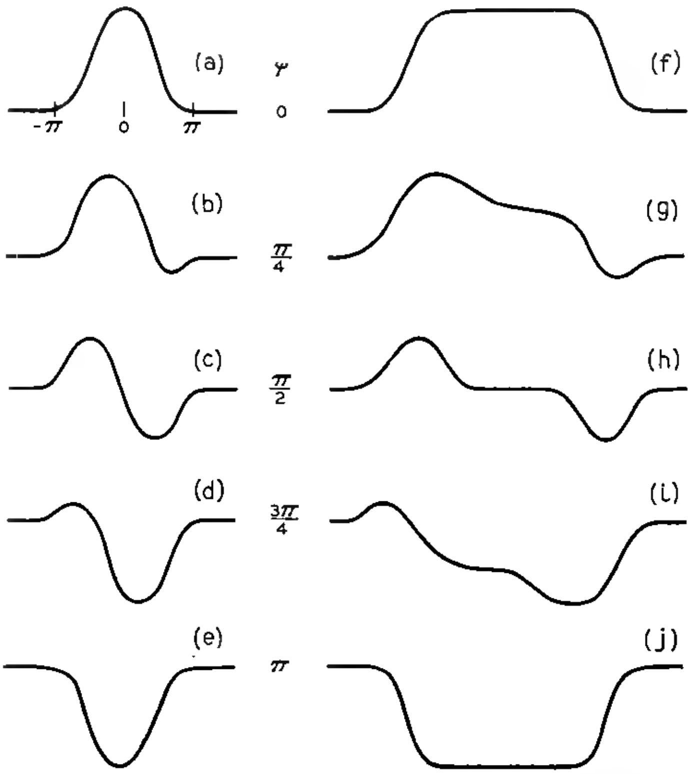

Fig. 7 — Distortion of baseband pulse signal in single sideband carrier facility.

The angle φ progresses through 0 to 2π per cycle of difference between the original and recovered frequencies. That is, if this difference is 2 cycles, then φ progresses from 0 to 2π twice each second.

The effect of the progression from 0 to π is illustrated in Figs. 7(a) to 7(e). When the phase is equal to π, as at 7(e), the signal is "upset"; i.e., marks are changed to spaces, and vice versa. When the phase is equal to $\pi/2$, the signal is effectively differentiated, as at 7(c).

In general, the signal as specified in (1) includes harmonics of Ω. An illustration of such a signal several dots long is shown in Fig. 7(f). As before, when $\varphi = \pi$, the complete signal is upset, as illustrated in Fig. 7(j). When $\varphi = \pi/2$, as at 7(h), the signal is more or less differentiated. The differentiation is not exact because the successive harmonics are not weighted according to order, as in true differentiation.

The distortions caused by the progression of φ are what make it difficult to recognize the signal at the receiving point. Some suggestions have been made for correcting the indication when the signal is upset, as in Figs. 7(e) and 7(j). It is more difficult, however, to take care of the intermediate cases, particularly 7(c) and 7(h).

APPENDIX II — EFFECT OF CHANNEL SUBDIVISION ON VULNERABILITY TO NOISE

There are three broad categories of noise to which the system may be exposed. These are:

1. Random noise which is not localized in time nor frequency.
2. Impulse noise which is highly localized in time but covers a broad frequency spectrum.
3. Single-frequency noise, which is highly localized in frequency but which lasts a significant time (or substantial number of bits).

We can assume two systems, A used as a single-frequency band, and B divided into ten channels. Correspondingly, therefore, signal pulses of unit duration over system A, are of 10 units duration in each channel of system B.

Random noise having uniform spectral distribution, and power W in each channel of system B, cumulates to power $10W$ in system A. Signal power P in each channel of system B cumulates to $10P$ for the total system. If signal power $10P$ is used in system A, the two systems are at a stand-off in signal-to-noise ratio for this type of noise.

In practice, the power capacity to handle the signals for system B must be made a few db higher than indicated by $10P$ to allow for occasional peaking caused during instants of unfavorable phasing among the various channels. Thus, the multichannel system B is really worse off, by that small amount, than system A. This may amount to some 3 or 4 db in ten or twelve channels.

There are occasions where crosstalk into other facilities sets the level permitted for the signal. It may be that concentration of the signal onto a single carrier aggravates this interference, as compared with that from a multi-carrier signal. In such cases system A may be penalized by a few db, as compared with system B.

Impulse noise shows correlation among the phases of its spectral components. Thus a noise pulse of voltage amplitude N in each channel of system B, cumulates to voltage amplitude $10N$ in system A or 20 db greater. On the other hand, a signal of amplitude S in each channel of system B, cumulates to an RMS voltage amplitude $\sqrt{10}\ S$, or 10 db greater, over the total system (plus a peaking correction which may be positive or negative as just mentioned). Thus, the single channel A system is at a disadvantage of 10 db, less the peaking adjustment, with respect to the B system.

A further adjustment may be needed, because a single noise peak that affects all 10 channels of B, or 10 bits, may affect only one bit of A. This adjustment depends upon the word grouping of bits which is used. It may be neglected if all the 10 bits of B are in the same word, and if, in one word, an error of 10 bits is effectively no worse than an error of 1 bit.

Single-frequency noise lies at the opposite extreme of the gamut from impulse noise. The vulnerability of a signal pulse to single-frequency noise varies according to the relationship between the frequencies of the noise and of the signal carrier in the utilized signal band.

The pattern of sensitivity to noise over an individual channel can be expected to be about the same for a narrow band as for a wide-band channel. Thus the pattern of sensitivities in the single channel of system A is repeated in each channel of system B on a 10 times finer frequency scale. The required S/N ratio in any one channel of B remains the same as that for A.

In system B each of the ten channels must put out only one tenth of the power of the single channel of system A (less the correction for peaking which as before may be positive or negative). Thus any one of the ten channels of B is 10 db (plus a peaking correction) more vulnerable to single frequency noise than the single channel of A.

It must be noted that there are occasional special circumstances where the single frequency noise may be persistent and steady. The multichannel system B may in such cases have an advantage in permitting the one channel affected to be dropped, and the others to be worked entirely free from this interference. This of course reduces the total bit rate.

To summarize the discussion in a general philosophical way, it can be said that there is advantage in multiplexing the signal in the manner that makes it as different as possible from the type of noise to which it is expected to be the most exposed. If the predominant noise is in short duration pulses, the most advantageous signal is in long duration pulses with frequency discrimination multiplex. If the noise is in longer duration single frequencies, the most advantageous signal is in very short pulses with time discrimination multiplex.

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